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EXAMINER	
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ART UNIT	PAPER NUMBER
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/697,810	BAXLEY ET AL.	
	Examiner	Art Unit	
	REDENTOR M. PASIA	2616	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _3-10,12-15 and 32-39_ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _3-10,12-15 and 32-39_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☒ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on _30 October 2003_ is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _10/30/2003_.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Specification

1. The abstract of the disclosure is objected to because the acronyms listed in the abstract does not have the proper terms that are related to them. It would be easier for a person of ordinary skill in the art to understand the abstract if each acronym has a related term it corresponds to at least once in the abstract. The examiner suggests adding the terms related to IVR, CACS, and MCU at least once (i.e interactive voice response (IVR).). Correction is required. See MPEP § 608.01(b).

Double Patenting

2. The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the "right to exclude" granted by a patent and to prevent possible harassment by multiple assignees. A nonstatutory obviousness-type double patenting rejection is appropriate where the conflicting claims are not identical, but at least one examined application claim is not patentably distinct from the reference claim(s) because the examined application claim is either anticipated by, or would have been obvious over, the reference claim(s). See, e.g., *In re Berg*, 140 F.3d 1428, 46 USPQ2d 1226 (Fed. Cir. 1998); *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422

F.2d 438, 164 USPQ 619 (CCPA 1970); and *In re Thorington*, 418 F.2d 528, 163 USPQ 644 (CCPA 1969).

A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) or 1.321(d) may be used to overcome an actual or provisional rejection based on a nonstatutory double patenting ground provided the conflicting application or patent either is shown to be commonly owned with this application, or claims an invention made as a result of activities undertaken within the scope of a joint research agreement.

Effective January 1, 1994, a registered attorney or agent of record may sign a terminal disclaimer. A terminal disclaimer signed by the assignee must fully comply with 37 CFR 3.73(b).

3. Claims 3-10, 12-15, 32-36 rejected on the ground of nonstatutory obviousness-type double patenting as being unpatentable over claims 1-9, 11-13, 22, 25-27 of U.S. Patent No. 6,646,997, hereinafter 997. Although the conflicting claims are not identical, they are not patentably distinct from each other because:

As to claims 6, 32, 33, and 34 of the application, claim 1 of 997 shows all of the elements recited in claims 6, 32, 33, and 34 of the application.

US 6,646,997	**Instant Application**
1. A method of large-scale fault-tolerant audio conferencing in a purely packet-switched audio conferencing system, said	6. A method of establishing an audio conference in an audio conferencing system, the method comprising:

method comprising the steps of: **placing a call** from an endpoint to a conference gatekeeper, said call indicating an audio conference;	**initiating a call** from an endpoint to said audio conferencing system, said call indicating said audio conference;
selecting in said conference allocation and control system a multiple control unit to host said audio conference **when said audio conference is inactive**; selecting in said conference allocation and control system a multiple control unit hosting said audio conference **when said audio conference is active;**	selecting, in a conference allocation and control system in said audio conferencing system, a multiple control unit to host said audio conference;
determining in said conference allocation and control system whether the call from said endpoint contains adequate information to establish said audio conference;	determining in said conference allocation and control system whether the call from said endpoint contains adequate information to establish said audio conference;
responding from said conference allocation and control system to said endpoint with routing instructions to an interactive voice response server when there is **other than said adequate**	responding from said conference allocation and control system to said endpoint with routing instructions to an interactive voice response server when there is **inadequate information** to

information to establish said audio conference;	establish said audio conference;
connecting said endpoint to said interactive voice response server when there is inadequate information to route said call;	connecting said endpoint to said interactive voice response server when there is inadequate information to route said call;
gathering in said interactive voice response server, after connecting said endpoint to said interactive voice response server, adequate information to establish said audio conference; and	gathering in said interactive voice response server, after connecting said endpoint to said interactive voice response server, said. adequate information to establish said audio conference; and
transferring said endpoint from said interactive voice response server to said selected multiple control unit after said interactive voice response server gathers said adequate information.	transferring said endpoint from said interactive voice response server to said selected multiple control unit after said interactive voice response server gathers said adequate information.
1. selecting in said conference allocation and control system a multiple control unit to host said audio conference when said audio conference is inactive;	32. The method of claim 6 wherein said selecting said multiple control unit comprises: selecting in said conference allocation and control system a first multiple control unit

	to host said audio conference when said audio conference is inactive.
1. selecting in said conference allocation and control system a multiple control unit hosting said audio conference when said audio conference is active;	33. The method of claim 6 wherein said selecting said multiple control unit comprises: selecting in said conference allocation and control system a second multiple control unit to host said audio conference when said audio conference is active.
1. responding from said conference allocation and control system to said endpoint with said queried routing instructions, said queried routing instructions indicating said selected multiple control unit;	34. The method of claim 6 further comprising: responding from said conference allocation and control system to said endpoint with queried routing instructions, said queried routing instructions indicating said selected multiple control unit.

In this case, when considering the claim limitation *"selecting, in a conference allocation and control system in said audio conferencing system, a multiple control unit to host said audio conference;"* in claim 6 of the instant application, it is noted that the said limitation is broader than *"selecting in said conference allocation and control*

system a multiple control unit to host said audio conference ***when said audio conference is inactive****; selecting in said conference allocation and control system a multiple control unit hosting said audio conference* ***when said audio conference is active;"*** as recited in claim 1 of Patent 997. Thus, it is noted that allowing this would result in an unjustified or improper timewise extension of the "right to exclude" granted by the patent. Claim 1 of Patent 997 as shown above, shows all the claim limitations set forth in Claims 32-34 (which depends on claim 1) of the instant application.

As to claims 7, 8, 9, 10, 12, 13, 14, 15, 3, 4, and 5 of the application, claims 2, 3, 4, 5, 6, 7, 8, 9, 11, 12, and 13 respectively, of 997 shows all of the elements recited in claims 7, 8, 9, 10, 12, 13, 14,15, 3, 4, and 5 of the application.

US 6,646,997	**Instant Application**
2. **A method of large-scale fault-tolerant audio conferencing in a purely packet-switched audio conferencing system**, said method comprising the steps of: placing a call from an endpoint to **a conference gatekeeper**, said call indicating an audio conference;	7. **A method for adding an additional endpoint to an audio conference in a purely packet-switched audio conferencing system**, said method comprising: placing a call from an endpoint to a **packet-switched conferencing system component**, said call indicating an audio conference;
selecting in said conference allocation and	selecting, in a conference allocation and

control system a multiple control unit to host said audio conference **when said audio conference is inactive**; selecting in said conference allocation and control system a multiple control unit hosting said audio conference **when said audio conference is active**;	control system in said audio conferencing system, a multiple control unit to host said audio conference;
initiating a call request from said selected multiple control unit to said **conference gatekeeper,** said call request indicating an additional endpoint	initiating a call request from said selected multiple control unit to said **packet-switched conferencing system component**, said call request indicating said additional endpoint;
returning a destination address to said **conference gatekeeper** from said gatekeeper cloud, said destination address corresponding to said additional endpoint	returning a destination address from said **packet-switched conferencing system component** to said selected multiple control unit, said destination address corresponding to said additional endpoint;
establishing a point-to-point call from said multiple control unit to said additional endpoint based on said destination address, thereby bringing said additional	establishing a point-to-point call from said multiple control unit to said additional endpoint based on said destination address, thereby bringing said additional

endpoint into said audio conference.	endpoint into said audio conference.
3. The method of claim 2 further supporting full service audio conferencing using a reservation system and a call agent.	8. The method of claim 7 further supporting full service audio conferencing using a reservation system and a call agent.
4. The method of claim 3 wherein the reservation system and the call agent are tightly integrated.	9. The method of claim 8 wherein the reservation system and the call agent are tightly integrated.
5. The method of claim 3 wherein the reservation system and the call agent are loosely integrated.	10. The method of claim 8 wherein the reservation system and the call agent are loosely integrated.
6. The method of claim 2 further including the step of dynamically routing an operator voice path to service multiple multiple control units.	12. The method of claim 7 further including dynamically routing an operator voice path to service multiple multiple control units.

7. The method of claim 2 further including the step of renegotiating the destination address of a voice path to move an audio conference participant from said selected multiple control unit to a second multiple control unit.	13. The method of claim 7 further including renegotiating the destination of a voice path to move an audio conference participant from said selected multiple control unit to a second multiple control unit.
8. The method of claim 2 further including the step of moving said audio conference from said selected multiple control unit to a second multiple control unit.	14. The method of claim 7further including moving said audio conference from said selected multiple control unit to a second multiple control unit.
9. The method of claim 2 further including the steps of: providing said audio conference to a streaming protocol server from said selected multiple control unit; connecting a passive participant to said streaming protocol server; and broadcasting said audio conference from said streaming protocol server to a passive	15. The method of claim 7 further comprising: providing said audio conference to a streaming protocol server from said selected multiple control unit; connecting a passive participant to said streaming protocol server; and broadcasting said audio conference from

participant.	said streaming protocol server to a said passive participant.

In this case, when considering the claim limitation ″**A method for adding an additional endpoint to an audio conference in a purely packet-switched audio conferencing system…**″ in claim 7 of the instant application, it is noted that the said limitation is broader than ″**A method of large-scale fault-tolerant audio conferencing in a purely packet-switched audio conferencing system..**″ as recited in claim 2 of Patent 997. Also, when considering the claim limitation ″**packet-switched conferencing system component** " in claim 7 of the instant application, it is noted that the said limitation is broader than ″**conference gatekeeper"** as recited in claim 2 of Patent 997. Thus, it is noted that allowing this would result in an unjustified or improper timewise extension of the "right to exclude" granted by the patent.

Claims 2, 3, 4, 5, 6, 7, 8, 9, 11, 12, and 13, of Patent 997, as shown above, shows all the claim limitations set forth in claims 7, 8, 9, 10, 12, 13, 14,15, 3, 4, and 5 (claims 8-15 and 3-5 depends on claim 1), respectively, of the instant application.

As to claims 35 and 36, of the application, claim 22 of 997 shows all of the elements recited in claims 35 and 36 of the application.

US 6,646,997	Instant Application
1. A method of large-scale fault-tolerant audio conferencing in a purely packet-switched audio conferencing system,	**32. A method of establishing an audio conference in an audio conferencing system**, the method comprising:

said method comprising the steps of: placing a call from an endpoint to a conference gatekeeper, said call indicating an audio conference;	initiating a call from an endpoint to said audio conferencing system, said call indicating said audio conference;
determining in said conference allocation and control system whether the call from said endpoint contains adequate information to establish said audio conference;	determining in a conference allocation and control system whether the call from said endpoint contains adequate information to establish said audio conference;
responding from said conference allocation and control system to said endpoint with routing instructions to an interactive voice response server when there is other than said adequate information to establish said audio conference;	responding from said conference allocation and control system to said endpoint with routing instructions to an interactive voice response server when there is inadequate information to establish said audio conference;
connecting said endpoint to said interactive voice response server when there is inadequate information to route said call;	connecting said endpoint to said interactive voice response server when there is inadequate information to route said call;
gathering in said interactive voice response server, after connecting said	gathering in said interactive voice response server, after connecting said

endpoint to said interactive voice response server, adequate information to establish said audio conference; and	endpoint to said interactive voice response server, said adequate information to establish said audio conference; and
transferring said endpoint from said interactive voice response server to said selected multiple control unit after said interactive voice response server gathers said adequate information.	transferring said endpoint from said interactive voice response server to said audio conference after said interactive voice response server gathers said adequate information.
selecting in said conference allocation and control system a multiple control unit to host said audio conference **when said audio conference is inactive**; selecting in said conference allocation and control system a multiple control unit hosting said audio conference **when said audio conference is active;**	36. The method of claim 35 further comprising: selecting, in said conference allocation and control system, a multiple control unit to host said audio conference.

In this case, when considering the claim limitation **"A method of establishing an audio conference in an audio conferencing system"** in claim 32 of the instant application, it is noted that the said limitation is broader than **"A method of large-scale fault-tolerant audio conferencing in a purely packet-switched audio conferencing system**, as recited in claim 1 of Patent 997. Also, when considering the claim limitation

"selecting, in a conference allocation and control system in said audio conferencing system, a multiple control unit to host said audio conference;" in claim 36 of the instant application, it is noted that the said limitation is broader than *"selecting in said conference allocation and control system a multiple control unit to host said audio conference* ***when said audio conference is inactive****; selecting in said conference allocation and control system a multiple control unit hosting said audio conference* ***when said audio conference is active;"*** as recited in claim 1 of Patent 997. Thus, it is noted that allowing this would result in an unjustified or improper timewise extension of the "right to exclude" granted by the patent.

Claim 22, of Patent 997 as shown above, shows all the claim limitations set forth in claims 35 and 36 (claim 36 depends on claim 35), of the instant application.

4. Claims 37, 38, 39 arerejected on the ground of nonstatutory obviousness-type double patenting as being unpatentable over claim 1 of U.S. Patent No. 6,646,997 in view of claims 25, 26, 27, of US Patent No. 6,646,997.

As to claim 37, claim 1 of Patent 997 shows all of the elements except the step of dynamically routing an operator voice path to service multiple multiple control units.

Claim 25 shows the above elements as shown below:

US 6,646,997	**Instant Application**
25. The method of claim 22 further including the step of dynamically routing an operator voice path to service multiple	37. The method of claim 36 further including dynamically routing an operator voice path to service multiple multiple

multiple control units.	control units.

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify claim 1 of Patent 997 to include the features of claim 25 of Patent 997 in order to have efficient control over a conferencing system.

As to claim 38, claim 1 of Patent 997 shows all of the elements except the step of renegotiating the destination of a voice path to move an audio conference participant from said selected multiple control unit to a second multiple control unit.

Claim 26 shows the above elements as shown below:

US 6,646,997	**Instant Application**
26. The method of claim 22 further including the step of renegotiating the destination of a voice path to move an audio conference participant from said selected multiple control unit to a second multiple control unit.	38. The method of claim 36 further including renegotiating the destination of a voice path to move an audio conference participant from said selected multiple control unit to a second multiple control unit.

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify claim 1 of Patent 997 to include the features of claim 26 of Patent 997 in order to have efficient control over a conferencing system.

As to claim 39, claim 1 of Patent 997 shows all of the elements except the step of moving said audio conference from said selected multiple control unit to a second multiple control unit.

Claim 27 shows the above elements as shown below:

US 6,646,997	Instant Application
27. The method of claim 22 further including the step of moving said audio conference from said selected multiple control unit to a second multiple control unit.	39. The method of claim 36 further including moving said audio conference from said selected multiple control unit to a second multiple control unit.

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify claim 1 of Patent 997 to include the features of claim 26 of Patent 997 in order to have efficient control over a conferencing system.

Thus, it is noted that allowing these, would result in an unjustified or improper timewise extension of the “right to exclude” granted by the patent.

Claim Rejections - 35 USC § 102

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

A person shall be entitled to a patent unless –

(e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States

only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

6. Claims 3, 7, 12, and 37 are rejected under 35 U.S.C. 102(e) as being anticipated by Detampel, Jr. et al (US 2001/0002927 A1; hereinafter Detampel).

As to claim 7, Detampel shows a method for adding an additional endpoint to an audio conference in a purely packet-switched audio conferencing system (Figure 1; abstract; method for setting up an on-demand conference call in a telecommunications system), said method comprising:

placing a call from an endpoint (figure 6, step 601) to a packet-switched conferencing system component (Figure 3, CACS 301), said call indicating an audio conference (Figure 6, step 601; Par. 0065, caller dials a unique on-demand conference number);

selecting, in a conference allocation and control system (Figure 1, CACS 103; Figure 3, CACS 301) in said audio conferencing system (figure 1, system 10), a multiple control unit (Figure 1; bridge server 101a-101n) to host said audio conference (Par. 0031, when an on-demand conference call request comes in, the CACS determines which bridge servers 101 have sufficient availability of ports to handle the on-demand conference call; Par. 0065; the steps take place as described above to select the bridge server 101 having enough ports available for the subscriber's maximum call);

initiating a call request from said selected multiple control unit (Figure 1, bridge 101a-n) to said packet-switched conferencing system component (Figure 3, Bridge Manager 310 in CACS 301), said call request indicating said additional endpoint (Par. 0075; as each participant joins the conference, a participant join message is sent via

BAPI module 303 and the bridge manager 310 (BAPI module 303 and bridge manager 310 are part of CACS).);

returning a destination address (Par. 0054; ONNET translation number) from said packet-switched conferencing system component to said selected multiple control unit (Par. 0054; The CACS call router module 302 receives a routing request from over network 109 from the SCP pair 105. The CACS 103 selects a bridge server 101 with enough available capacity to handle the maximum number of conference participants allowed by the service (e.g., 20), allocates the capacity, and returns routing instructions in terms of a POTS or ONNET translation number through the SCP pair 105. Each bridge server 101 would have a unique POTS or ONNET translation number for every simultaneous conference allowed on the bridge.), said destination address corresponding to said additional endpoint (Par. 0054; for example, if the on-demand service had a maximum participant capacity of 20, each bridge server would need at least 12 (240 ports/20 participants per conference) unique POTS or ONNET translation numbers).

establishing a point-to-point call (Examiner notes that this claim limitation interpreted to be the same as being “a connection”) from said multiple control unit to said additional endpoint based on said destination address thereby bringing said additional endpoint into said audio conference (Par. 0054; the CACS 103 would note which translation number is currently "assigned" to which N00 number (unique number dialed by participant) so that a translation number can be tied to the dialed N00 number which can be tied to a participant which can be tied to the currently authorized

participant/subscriber passcode. The SCP pair 105 encapsulates the routing instructions with an SS7 TCAP message and returns appropriate routing instructions via the SS7 network 106 to the originating service point 402. The on-demand call is routed via the PSTN 102 to the selected bridge server 101.).

As to claim 3, Detampel shows that the step of placing a call, links said endpoint (figure 1, user in network 106, 102; Figure 4, user 401-n) Figure 2 to said packet-switched conferencing system component (Figures 1, 4, CACS 103) through said packet-switched audio conferencing system (Figures 1, 4, 6; Par. 0054).

As to claim 12, Detampel further shows the step of dynamically routing an operator voice path to service (Examiner interprets this claim limitation as being the same as having an operator being able to service/handle components/servers in a packet switched network; Par. 0042, shows the Operator Interface module 305 is the application program interface to the operator/maintenance stations 107, and handles operator request queue management, registration for operator-monitored bridge events, and operator updates to the subscriber database 104; Figure 6, Par. 0070, 0074, 0075; shows the operator functions when an invalid passcode/PIN was supplied, however, for example purposes, the operator station is shown to interact with bridge 101.; Par. 0029; shows operator/maintenance stations 107 is connected to CACS through network 109 to provide operator interaction with system 10, that further includes multiple bridge servers 101a-n) multiple multiple control units (Figure 1, bridge servers 101a-n).

As to claim 37, this claim is rejected using the same reasoning set forth in the rejection of claim 12.

Claim Rejections - 35 USC § 103

7. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

8. Claims 6, 32-36 are rejected under 35 U.S.C. 103(a) as being unpatentable over Detampel, Jr. et al (US 2001/0002927 A1; hereinafter Detampel) in view of Petrunka et al. (US 6,122,364; hereinafter Petrunka).

As to claim 6, Detampel shows a method of establishing an audio conference in an audio conferencing system (abstract; method for setting up an on-demand conference call in a telecommunications system), the method comprising:

initiating a call from an endpoint to said audio conferencing system (Figure 6, step 601, caller dials), said call indicating said audio conference (Figure 6, step 601; Par. 0065, caller dials a unique on-demand conference number);

selecting, in a conference allocation and control system (Figure 1, conference allocation and control system, CACS 103) in said audio conferencing system (figure 1, system 10), a multiple control unit (Figure 1; bridge server 101a-101n) to host said audio conference (Par. 0031, when an on-demand conference call request comes in, the CACS 103 determines which bridge servers 101 have sufficient availability of ports to handle the on-demand conference call; Par. 0065; the steps take place as described

above to select the bridge server 101 having enough ports available for the subscriber's maximum call);

determining in said conference allocation and control system whether the call from said endpoint contains adequate information to establish said audio conference (Figure 6, steps 605-607, 617-619, prompting by CACS for passcode/PIN from user and validating the passcode/PIN);

responding from conference allocation and control system to said endpoint with routing instructions when there is inadequate information to establish said audio conference (Figure 6, steps 605-607; Par. 0070, 0075, after three incorrect attempts to enter the passcode/PIN, the caller is routed (not shown in FIG. 6) to an operator station 107 or a wrong passcode/PIN recording is played). Detampel further shows adequate information to establish said audio conference (Figure 6, 605-607) and said selected multiple control unit (Figure 1, bridge server 101a-n; Par. 0031, when an on-demand conference call request comes in, the CACS 103 determines which bridge servers 101 have sufficient availability of ports to handle the on-demand conference call). However, Detampel does not show an interactive voice response server and the steps of: connecting said endpoint to said interactive voice response server when there is inadequate information to route said call; gathering in said interactive voice response server, after connecting said endpoint to said interactive voice response server, said adequate information to establish said audio conference; and transferring said endpoint from said interactive voice response server to said selected multiple control unit after said interactive voice response server gathers said adequate information.

Petrunka shows an interactive voice response server (Figure 1, IVR 1130) and the steps of: connecting said endpoint to said interactive voice response server when there is inadequate information to route said call (Figure 4, steps 4010-4040); gathering in said interactive voice response server (Figure 4, step 4050), after connecting said endpoint to said interactive voice response server (Figure 4, steps 4010-4040), said adequate information (Figure 4, step 4050, additional information); and transferring said endpoint from said interactive voice response server after said interactive voice response server gathers said adequate information (Figure 4, steps 4050 to 4070; shows that when additional information is verified, the high level switch (which includes the IVR and the automatic call distribution (ACD) server) directs the call to VoIP server). It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Detampel to include the features, as shown above, of Detampel, in order to eliminate the requirement for an additional data path, and/or special circuitry (col. 1, lines 63-65).

As to claim 32, modified Detampel shows that said selecting said multiple control unit comprises: selecting in said conference allocation and control system a first multiple control unit to host said audio conference (Detampel: Figure 6, steps 601-602; Par. 0065; on-demand conference begins 600 when the caller dials in the unique on-demand number 601. The steps take place as described above to select the bridge server 101 having enough ports available for the subscriber's maximum call.) when said audio conference is inactive (Detampel: Par. 0065; a decision is made as to whether the conference has begun 602).

As to claim 33, modified Detampel shows the step of selecting in said conference allocation and control system a second multiple control unit to host said audio conference when said audio conference is active (Detampel: Figure 6, shows that when a user dials the designated number, step 602 checks if the conference has begun and allows the caller to connect to the conference if the caller provides adequate information; Figure 5, shows the method of selecting which bridge (Detampel: Figure 1, bridge servers 101a-n) would have enough resources to handle a caller/subscriber; Par. 0054, 0057-0060).

As to claim 34, modified Detampel shows the steps of responding from said conference allocation and control system to said endpoint with queried routing instructions, said queried routing instructions indicating said selected multiple control unit (Detampel; Par. 0054; The CACS call router module 302 receives a routing request from over network 109 from the SCP pair 105. The CACS 103 selects a bridge server 101 with enough available capacity to handle the maximum number of conference participants allowed by the service (e.g., 20), allocates the capacity, and returns routing instructions in terms of a POTS or ONNET translation number through the SCP pair 105. Each bridge server 101 would have a unique POTS or ONNET translation number for every simultaneous conference allowed on the bridge.).

As to claim 35, this claim is rejected using the same reasoning set forth in the rejection of claim 6.

As to claim 36, modified Detampel shows the step of selecting, in a conference allocation and control system (Detampel: Figure 1, conference allocation and control

system, CACS 103) in said audio conferencing system (Detampel: figure 1, system 10), a multiple control unit (Detampel: Figure 1; bridge server 101a-101n) to host said audio conference (Detampel: Par. 0031, when an on-demand conference call request comes in, the CACS 103 determines which bridge servers 101 have sufficient availability of ports to handle the on-demand conference call; Par. 0065; the steps take place as described above to select the bridge server 101 having enough ports available for the subscriber's maximum call).

9. Claims 4-5 are rejected under 35 U.S.C. 103(a) as being unpatentable over Detampel, Jr. et al (US 2001/0002927 A1; hereinafter Detampel) in view of Thomas (US 6,421,339 B1; hereinafter Thomas).

As to claim 4, Detampel shows all of the elements except a location found signal indicating the selected multiple control unit.

Thomas shows a location found signal indicating the selected multiple control unit (Figure 3, col. 5, lines 25-30; gatekeeper GK 14 may screen or otherwise filter the data received in the LCF message from GK 44 and then send a LCF to the requester or calling endpoint. As will be obvious to network designers, the data returned to the calling party may be limited so that calls must be routed through the home gatekeeper rather than giving the calling endpoint enough data to place a call directly to a roaming user). It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Detampel to include the features of Thomas, as

shown above, in order to allow the gatekeeper to monitor the contents of all call received by given users (col. 5, lines 32-33).

As to claim 5, Detampel shows all of the elements except a location request signal.

Thomas shows a location request signal (Figure 3, LRQ). It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Detampel to include the features of Thomas, as shown above, in order to allow the gatekeeper to monitor the contents of all call received by given users (col. 5, lines 32-33).

10. Claims 8-10 are rejected under 35 U.S.C. 103(a) as being unpatentable over Detampel, Jr. et al (US 2001/0002927 A1; hereinafter Detampel) in view of Jurkevics et al. (US 5,978,463; hereinafter Jurkevics).

As to claim 8, Detampel shows all of the elements except supporting full service audio conferencing using a reservation system and a call agent.

Jurkevics shows full service audio conferencing (Figures 2-4; abstract, audio conferencing system) using a reservation system (Figure 4, Autoscheduler 28) and a call agent (Figure 1, client 10, Figure 4, Client program 20 running on Client 10). It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Detampel to include the features of Jurkevics, as shown above, in order to provide a substantially less labor intensive approach in audio conference scheduling (col. 3, lines 16-20).

As to claim 9, modified Detampel shows that the reservation system and the call agent are tightly integrated (Jurkevics: Figure 4-5, shows the integration of the automatic scheduling system with the client program in scheduling a conference; col. 5, lines 33-48; shows different levels of service, unattended service (no agent attending the audio conference), standard level, and premiere level).

As to claim 10, modified Detampel shows that the reservation system and the call agent are loosely integrated (Jurkevics: Figure 4-5, shows the integration of the automatic scheduling system with the client program in scheduling a conference; col. 5, lines 33-48; shows different levels of service, unattended service (no agent attending the audio conference), standard level, and premiere level).

11. Claims 13-14, 38-39 are rejected under 35 U.S.C. 103(a) as being unpatentable over Detampel, Jr. et al (US 2001/0002927 A1; hereinafter Detampel) in view of Semaan (US 5,680,392; hereinafter Semaan).

As to claim 13, Detampel shows all of the elements except the step of renegotiating the destination of a voice path to move an audio conference participant from said selected multiple control unit to a second multiple control unit.

Semaan shows the step of renegotiating the destination of a voice path to move an audio conference participant from said selected multiple control unit to a second multiple control unit (Figure 2, 5; col. 11, lines 18-25; shows that if a user should wish to establish a conference with conferees who would be handled by the reservation controller of another domain, the bridge controller would pass the reservation request

information onto the reservation request channel of the other reservation domain so that the appropriate reservation controller in the other domain could address the request; Figure 2 and 5, shows that each reservation controller is related to an MCU). It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Detampel to include the features of Semaan, as shown above, in order to provide the possibility of allowing different MCUs and reservation controllers (of different companies), to interact with each other and share information regarding requests for reservations (col. 5, lines 29-37).

As to claim 14, Detampel shows all of the elements except the step of moving said audio conference from said selected multiple control unit to a second multiple control unit.

Semaan shows the step of moving said audio conference (Figure 2, 5; col. 11, lines 18-25; shows that if a user should wish to establish a conference with conferees who would be handled by the reservation controller of another domain, the bridge controller would pass the reservation request information onto the reservation request channel of the other reservation domain so that the appropriate reservation controller in the other domain could address the request; Figure 2 and 5, shows that each reservation controller is related to an MCU) from said selected multiple control unit to a second multiple control unit (Examiner notes that there is a change in reservation controllers, there is also a change in MCUs). It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Detampel to include the features of Semaan, as shown above, in order to provide the possibility of

allowing different MCUs and reservation controllers (of different companies), to interact with each other and share information regarding requests for reservations (col. 5, lines 29-37).

As to claims 38 and 39, these claims are rejected using the same reasoning set forth in the rejection of claims 13 and 14, respectively.

12. Claim 15 rejected under 35 U.S.C. 103(a) as being unpatentable over Detampel, Jr. et al (US 2001/0002927 A1; hereinafter Detampel) in view of Semaan (US 5,680,392; hereinafter Semaan) in further view of Rosenberg et al. (US 2005/0165894 A1; hereinafter Rosenberg).

As to claim 15, Detampel shows selected multiple control unit (Figure 1, bridge server 101a-n). However, Detampel does not explicitly show the steps of providing said audio conference to a streaming protocol server from said selected multiple control unit ; connecting a passive participant to said streaming protocol server; and broadcasting said audio conference from said streaming protocol server to a said passive participant.

Semaan shows the steps of providing said audio conference to a reservation controller from said selected multiple control unit (Figure 2, 5; col. 11, lines 18-25; shows that if a user should wish to establish a conference with conferees who would be handled by the reservation controller of another domain, the bridge controller would pass the reservation request information onto the reservation request channel of the other reservation domain so that the appropriate reservation controller in the other

domain could address the request; Figure 2 and 5, shows that each reservation controller is related to an MCU); connecting a passive participant to said reservation controller (col. 11, lines 18-25; col. 5, lines 20-29; if users 112c, 112e, 112f, 112g, 112h, and 112j should wish to participate in a multimedia conference, the services of the four different MCUs 126a-126d will be required. Thus, the two reservation controllers 130a, 130b must be contacted to reserve appropriate access and processing of the MCUs.); and broadcasting said audio conference from said reservation controller to a said passive participant (col. 8, line 65 to col. 9, line 9; shows that the conference mode includes broadcast monologue and broadcast dialogue). It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Detampel to include the features of Semaan, as shown above, in order to provide the possibility of allowing different MCUs and reservation controllers (of different companies), to interact with each other and share information regarding requests for reservations (col. 5, lines 29-37). To reiterate, Semaan shows that the conference is held by the respective reservation controller (along with proper MCU), however, he does not explicitly show that the conference is being provided to a streaming protocol server.

Rosenberg shows a streaming protocol server (Par. 0168; a conference participant can invite a SIP-speaking RTSP server into an existing conference, so as to appear as just another conference participant. Alternatively, for multicast conferences, an RTSP server can simply be given the same session description as was used for invitations). It would have been obvious to one of ordinary skill in the art at the time of the invention to further modify the method of modified Detampel to include the features

of Rosenberg, as shown above, in order to provide one or more relatively advanced telephony services (Par. 0015).

Conclusion

13. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Kwak et al. (US 6,457,043 B1);

Yegoshin (US 2002/0093942 A1);

Chang (US 6,304,648 B1);

Andrews et al. (US 5,333,133);

Bruno (US 5,784,561);

Polcyn (US 6,195,357 B1).

Any inquiry concerning this communication or earlier communications from the examiner should be directed to REDENTOR M. PASIA whose telephone number is (571)272-9745. The examiner can normally be reached on M-F 7:30am to 4:00pm EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Aung Moe can be reached on (571)272-7314. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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